CLAIMS

1. A method of processing signals received from an array of sensors comprising the steps of sampling and digitally converting the received signals and processing the digitally converted signals to provide an output signal, the processing including filtering the signals using a first adaptive filter arranged to enhance a target signal of the digitally converted signals and a second adaptive filter arranged to suppress an unwanted signal of the digitally converted signals and processing the filtered signals in the frequency domain to suppress the unwanted signal further.

2. A method as claimed in claim 1 further comprising the step of determining a signal energy from the signals and determining a noise energy from the signal energy.

3. A method as claimed in claim 2 wherein the signal energy is determined by buffering N/2 samples of the digitized signal into a shift register to form a signal vector of the following form:

$$X_r = \begin{bmatrix} X(0) \\ X(1) \\ \cdot \\ \cdot \\ \cdot \\ X(J-1) \end{bmatrix}$$

Where J = N/2; and estimating the signal energy using the following equation:

$$E_r = \frac{1}{J-2} \sum_{i=1}^{J-2} X(i)^2 - X(i+1)\, X(i-1)$$

where E_r is the signal energy.

4. A method as claimed in claim 2 or claim 3 wherein the noise energy is determined by measuring the signal energy E_r of blocks of the digitally converted signals and calculating the noise nergy E_n in accordance with:

$$E_n^{K+1} = \alpha E_n^K + (1-\alpha) E_r^{K+1}$$

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Where the superscript K is the block number and α is an empirically chosen weight.

5. A method as claimed in any one of claims 2 to 4 further comprising the step of determining a noise threshold from the noise energy and updating the noise energy and noise threshold when the signal energy is below the noise threshold.

6. A method as claimed in claim 5 further comprising the step of determining if a target signal is present by comparing the signal energy to a signal threshold.

7. A method as claimed in claim 6 further comprising the step of determining the signal threshold from the noise energy and updating the signal threshold when the signal energy is below the noise threshold.

8. A method as claimed in any one of claims 5 to 7 wherein the noise threshold T_{n1} is determined in accordance with:

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$$T_{n1} = \delta_1 E_n$$

Where δ_1 is an empirically chosen value.

9. A method as claimed in claim 6 or claim 7 wherein the signal threshold T_{n2} is determined in accordance with:

$$T_{n2} = \delta_2 E_n$$

Where δ_2 is an empirically chosen value.

10. A method as claimed in any one of the preceding claims further comprising the step of determining the direction of arrival of the target signal.

11. A method as claimed in claim 10 further comprising the step of processing the

signals from two spaced sensors of the array with a third adaptive filter to determine said direction of arrival.

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12. A method as claimed in claim 10 or claim 11 further comprising the step of treating the signal as an unwanted signal if the signal has not impinged on the array from within a selected angular range.

13. A method as claimed in any one of the preceding claims further comprising the step of calculating a measure of the cross-correlation of signals from two spaced sensors of the array and treating the signal as an unwanted signal if the degree of cross correlation is less than a selected value.

14. A method as claimed in claim 11 further comprising the step of calculating a measure of reverberation of the signal from filter weights of the first and third adaptive filters.

15. A method as claimed in claim 14 wherein the reverberation measure C_{rv} is calculated in accordance with

$$C_{rv} = \frac{W_{td}^{T}\,W_{su}}{\|W_{td}\|\,\|W_{su}\|}$$

where T denotes the transpose of a vector, W_{su} is the filter coefficient of the first filter and W_{td} is the filter coefficient of the third filter.

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16. A method as claimed in claim 14 or claim 15 further comprising the step of treating the signal as an unwanted signal if the reverberation measure indictates a degree of reverberation in excess of a selected value.

17. A method as claimed in any one of the preceding claims further comprising the step of controlling the operation of the first filter to perform adaptive filtering only when

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a said target signal is deemed to be present.

18. A method as claimed in any one of the preceding claims wherein the first adaptive filter has a plurality of channels receiving as input the digitized signals and providing as output a sum and at least one difference signal, the difference signal channels including filter elements having corresponding filter weights.

19. A method as claimed in claim 18 further comprising the step of calculating a ratio of the energy in the sum and difference channels.

20. A method as claimed in claim 19 further comprising the step of treating the signal as including a said target signal if the ratio indicates that the energy in the sum channel is greater than the energy in the difference channels by more than a selected factor.

21. A method as claimed in claim 20 further comprising the step of treating the signal as including a said target signal only if the signal energy exceeds a threshold.

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22. A method as claimed in any one of the preceding claims further comprising the step of controlling the operation of the second filter to perform adaptive filtering only when a said target signal is deemed not to be present.

23. A method as claimed in any one of the preceding claims wherein the first adaptive filter has a plurality of channels receiving input signals from the first adaptive filter and providing as output a sum signal received from the first adaptive filter, an error signal and at least one difference signal, the difference signal channels including further filter elements having corresponding further filter weights.

24. A method as claimed in claim 23 further comprising the step of scaling the further filter weights if the norms of the further filter weights exceed a threshold.

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25. A method as claimed in claim 23 or claim 24 further comprising the step of

combining the sum signal and the error signal to form a single signal S(t) of the form:

$$S(t) = W_1 S_c(t) + W_2 e_c(t)$$

where $S_c(t)$ is the sum signal at time t, $e_c(t)$ is the error signal at time t and W_1 and W_2 are weight values.

26. A method as claimed in claim 25 further comprising the step of combining the difference signals to form a single signal.

27. A method as claimed in claim 25 or claim 26 further comprising the step of applying a Hanning window to the single signals.

28. A method as claimed in any one of the preceding claims further comprising the step of transforming the filtered signals into two frequency domain signals a desired signal S_f and an interference signal I_f, , processing the transformed signals to provide a gain for the desired signal and transforming the gain modified desired signal back to the time domain to provide an output.

29. A method as claimed in claim 28 wherein the processing step comprises the step of forming spectra for the frequency domain signals.

30. A method as claimed in claim 29 whererin the spectra are modified spectra P_s, P_i of the desired signal and the interference signal of the form:

$$P_s = |Real(S_f)| + |Imag(S_f)| + F(S_f)*r_s$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + F(I_f)*r_i$$

Where "Real" and "Imag" refer to taking the absolute values of the real and imaginary parts, r_s and r_i are scalars and $F(S_f)$ and $F(I_f)$ denotes a function of S_f and I_f respectively.

31. A method as claimed in claim 30 wherein the function is a power function.

32. A method as claimed in claim 31 wherein the spectra are of the form, where "Conj" denotes the complex conjugate:

$$P_i = |Real(I_f)| + |Imag(I_f)| + (I_f * conj(I_f)) * r_i$$

$$P_s = |Real(S_f)| + |Imag(S_f)| + (S_f * conj(S_f)) * r_s$$

33. A method as claimed in claim 30 wherein the function is a multiplication function

34. A method as claimed in claim 33 wherein the spectra are of the form:

$$P_s = |Real(S_f)| + |Imag(S_f)| + |Real(S_f)| * |Imag(S_f)| * r_s$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + |Real(I_f)| * |Imag(I_f)| * r_i$$

35. A method as claimed in any one of claims 29 to 34 wherein the processing step includes the step of warping the signal and interference spectra into a Bark scale to form corresponding signal and interference Bark spectra.

36. A method as claimed in claim 35 wherein the processing step further includes the step of calculating a system noise Bark spectrum.

37. A method as claimed in claim 36 further comprising the step of combining the interference Bark spectrum and the system noise Bark spectrum to form a combined noise Bark spectrum.

38. A method as claimed in claim 37 wherein the combined noise Bark spectrum B_y is of the following form:

$$B_y = \Omega_1 B_i + \Omega_2 B_n$$

where Ω_1 and Ω_2 are weighting values B_i is the interference Bark spectrum and B_n is the system noise Bark spectrum.

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39. A method as claimed in any one of claims 29 to 38 further comprising the step of calculating a signal to noise ratio from the spectra and deriving the gain from the signal to noise ratio.

40. A method as claimed in claim 39 further comprising the step of modifying the signal to noise ratio with a scaling factor which gradually changes from a first value at onset of the signal to a second value at which the scaling factor remains as the signal continues, until the signal ceases at which time the scaling factor is rest to the first value.

41. A method as claimed in claim 40 wherein the scaling factor changes in a plurality of steps.

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42. A method as claimed in claim 40 or claim 41 wherein the scaling factor changes exponentially.

43. A method of calculating a spectrum from a coupled signal comprising the steps of:

1) deriving a target signal component S and an interference signal component I from the coupled signal;

2) transforming the target and interference signal components into respective frequency domain equivalents F(S) and F(I);and

3) constructing the spectrum P(S) and P(I) of at least one equivalent in accordance with:

$$P(S)=|Real(F(S))| + |Imag(F(S))| + G[F(S)]*R(S)$$

$$P(I)=|Real(F(I))| + |Imag(F(I))| + G[F(I)]*R(I)$$

Where Real and Imag refer to taking the absolute value of the real or imaginary part of the frequency domain equivalent R(S), R(I) are scalar adjustment factors and G[F(S)] and G[F(I)] are functions of F(S) and F(I) respectively.

44. A method of calculating a reverberation coefficient from a plurality of signals received from respective sensors in respective signal channels of a sensor array comprising the steps of:

1) calculating a correlation time delay between signals from a reference one of the channels and another one of the channels using an adaptive filter;

2) performing adaptive filtering, using a second adaptive filter, on the received signals;and

3) calculating a reverberation coefficient from the filter coefficients of the first and second filters.

45. A method of signal processing of a signal having wanted and unwanted components comprising the steps of:

1) processing the signal in the time domain with at least one adaptive filter to enhance the wanted signal and/or reduce the unwanted signal,

2) transforming the thus processed signal to the frequency domain; and

3) performing at least one unwanted signal reduction process in the frequency domain.

46. Apparatus for performing the method of any one of the preceding claims.